Feasibility of Acoustic Sorting for Black Materials in Solid Waste Processing

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Abstract

Municipal and Industrial Solid Wastes are generally collected as mixtures of different solid materials. In solid waste treatment plants the waste gets crushed, classified and sorted. Among these processes the sorting is the determining step in which materials with the same recycling attributes are concentrated and cleaned from impurities. During the last decade sensor based sorting like 3D cameras and Near – Infra – Red (NIR) sorting made a huge technical improvement and enabled its industrial implementation. One of the still remaining problem lies within the sorting of black materials like plastics and rubbers which are very difficult to sort by using conventional visual sensors. This results from absorption of NIR emissions. The black materials have different structures and acoustic emissions when an impact is given. By using the acquisition and analysis of the acoustic signals from the impact in frequency-domain, the characteristics and features of different materials can be extracted and then transferred to the sorting system as sorting criteria. The key device of acoustic signal acquirement and analysis is the Data Acquisition (DAQ) card. With the high development of signal processing technologies, the capacity, compatibility, stability and flexibility of a soundcard is already adequate for industrial measurements and its price is much lower than the professional DAQ cards. The Signal acquirement and analysis systems which are discussed in this paper are therefore based on computer soundcards.

Keywords

Sensor based sorting, Impact acoustic, Spectrum analysis, LabVIEW

1 Introduction

Each year billions of tons of solid waste are generated through human activities in the world. All of them need to be treated in order to avoid pollution and other hazards which probably happen. In addition to that, most of the solid waste can be utilized as a source for secondary raw materials, like metals, plastics, glass, paper, old tires, etc. By recycling and utilizing solid waste materials a lot of energy, resources and raw materials can be saved which leads to improvement of local and global environmental.

Waste is always collected and fed into waste treatment plants as a mixture of different solid materials. In waste treatment plants the mixture gets crushed, classified and sorted. Among these processes the sorting is the determining step in which materials with the same recycling attributes are concentrated and cleaned from impurities.

The sorting technologies can be divided into direct and indirect separation processes. In direct processes there are different selective interactions between the characteristics of single particles and the corresponding force field of the separator. One example for the direct process is the eddy current separation.

However, the colour, texture or volume of each particle could also be considered as sorting criterion but there is no sufficient force field by which the corresponding particles get sorted from a mixture stream. According to this a separation based on recognition and mechanical sorting is necessary and defined as indirect sorting process. For example in the manual sorting processes, the characteristics of the particles are detected by the human eye and the material groups are organized in the human brain. After the detection, information which has been processed in the human brain is used to initialize the sorting operation with the hand. Similarly the methodologies which are defined as sensor based sorting processes follow the same basic principles like hand sorting. Instead of human senses different technical sensors are developed and implemented [7]. Examples are cameras, microphones and even some senses that cannot be processed by humans such as NIR and microwave sensors. In the last decade the sensor based sorting by using 3D cameras and NIR sensors made a huge technical improvement and enabled its industrial implementation. One of the still remaining problems lies within the sorting of black materials like plastics and rubbers which are very difficult to sort by using conventional visual sensors. This results from absorption of Near–Infra–Red (NIR) emissions. The black materials have different structures and acoustic emissions when an impact is given.

Although the black materials have the same visual characteristics, they still have some different characteristics like the acoustic emissions by impaction on solid surface. According to our experiences, the acoustic signal of plastic, rubber, and mineral materials colliding with a solid surface can be easily distinguished by hearing. The modern acoustic sensors like microphones are much more sensitive than the human ear and together with rapid signal processing technologies the acoustic sorting of black solid waste in industrial scale can be realized. Some facilities using this system have already been developed and employed, such as the processing system for nut and wheat kernels.

Different from the metals, most of the acoustic emissions of plastics and rubbers concentrate in the auditory threshold i.e. the frequency range of 20 – 20,000 Hz. In this range the PC soundcard is a perfect signal acquisition and analysis system. With the high development of signal processing technologies, the capacity, compatibility, stability and flexibility of computer soundcards are already adequate for industrial measurements and their price is much lower than the one of professional DAQ cards. Through the installation of several soundcards in one PC, a multi – channel signal acquisition and analysis system can be established. The signals can be processed and analyzed by virtual equipment with the corresponding software LabVIEW. The feature “extraction of acoustic signals” is based on frequency domain analysis.

This paper summarizes the results of experiments, in which the impact behaviours of several kinds of plastics have been studied with self-built acoustical equipment by the Department of Processing and Recycling (IAR) of the RWTH Aachen University. The frequency – domain – based analysis are used to process the signals and the spectral features are introduced to recognize the different materials. Nearly all of the tested materials have their own spectrum according to different particle size except some abnormal impacts. The feature “information of acoustic signals” is sufficient and available for the sorting criterion.

2 Preparation of the Experiment

2.1 Construction of the Experimental System

In order to acquire the acoustic emissions of different materials, individual sample particles of each material are designed to fall from a defined height and then impact on a thick stone plate. The impact system is sealed in an empty medium – density fiberboard (MDF) case whose inner surface is covered by sponge material in order to avoid the influence of ambient noises.

The MDF case consists of 4 elements. The height of the case can be adjusted to be 600 mm, 750mm or 900mm for different falling heights. The outer dimension of this equipment is 300 mm in width and 300 mm in length, the thickness of the MDF plates is 15 mm. The inner dimension is about 270 mm in width and 270mm in length. The impact stone plate is made of nature stone for foot paths with the dimension of 200 mm in length, 200 mm in width and the thickness of 30 mm. The stone plate is placed in a steel bracket which is set in a 45° angle inside the system. The acoustic signals of impacts are acquired by the acoustic sensor i.e. the microphone which is placed at the ceiling of the inner space [1].

The reason for the selection of the impact plate material is the excellent stability of the stone plate and its vibration characteristics towards plastics and rubber. A metal impact plate has similar acoustic emissions and a wooden plate would be deformed and worn off in short period of time.

The sketch of the case is illustrated in figure 1 [1] and the installation of the whole equipment and the impact plate are illustrated in figure 2 and 3 [1].

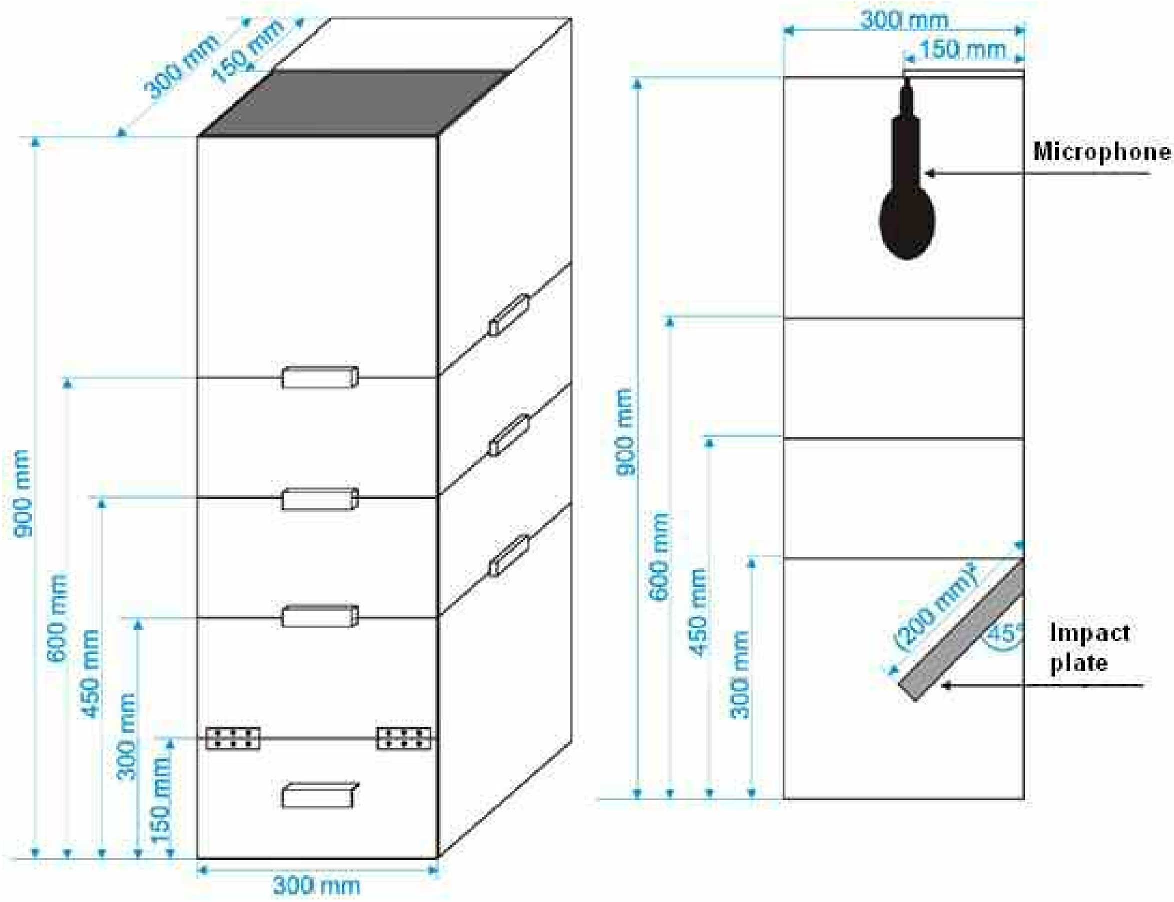

Figure 1 The Sketch of the experiment equipment for impact acoustics

Figure 2 Complete Equipment and the installation of microphone and impact plate

Figure 3 Impact plate for experiment (left: before experiment, right: after the experiment)

2.2 Demand of Software and Device

The original impact signals are acquired by a microphone and transferred to the computer. There they are saved and analyzed by the soundcard and LabVIEW. The professional DAQ cards which are available on the market like the device from National Instrument (NI) are expensive and the same functions can also be realized by standard soundcards of computer. The A/D and D/A capacity of soundcard are already adequate for industrial measurement and their prices are much lower than those of professional devices. Normally the precision of 16 bits soundcards is better than the 12 bits DAQ cards. The soundcards transfer the data by direct memory access (DMA) technology and result in massive reduction of the CPU occupation. The PCI bus technology allows the high speed data communication between soundcard and system and made the online analysis and real time manipulation by using virtual instruments possible. The key performances of soundcard are sampling rate and resolution. Currently the maximum sampling rate of a soundcard can reach up to 96 kHz, and the resolution can reach up to 32 bits and the maximum SNR reaches up to 114 dB. Normally, the function and operability of an external soundcard is better than the same of devices which are integrated in the common main – boards. Software updates are also more convenient for the external devices. The soundcard which was implemented in this project is an external soundcard, type of AUREON 5.1 USB MKII. The performances of this model are:

1. Sampling rate: 32, 44.1 and 48 kHz
2. Band width: 0 – 22.05 kHz
3. Resolution (number of samples pro sampling): 16 bit
4. Input connection: Line – in / Mic – in
5. Output connection: USB 1.1 / 2.0

The virtual instrument software which was used for data acquisition and analysis is LabVIEW (version 8.5). LabVIEW is designed for virtual instrument development. It is an advanced platform for industrial testing, measurement and manipulation. It includes almost all the common signal processing functions and lots of advanced signal processing toolkits. The virtual instrument (VI) program can be easily integrated with other hardware, Ethernet, BUS communicator and common databases.

2.3 The Method of Signal Processing and Analysis

The characteristic wave of acoustic signals can be expressed longitudinal wave which comes from a vibration source. The sound wave is transferred by media (air, water, iron, etc) as the continuous variation of amplitude, frequency, phase and some physicals. All the variations are detected by microphone and then converted to analog signals. The analog signals are converted to digital signals by the soundcard and then saved on the computer. Like the communication signal processing the analysis of acoustic signals can also be operated by frequency – domain analysis which is based on the Fast Fourier Transformation (FFT) method, hence the main feature of sound signals is the energy distribution according to frequencies. Standard methods for feature extraction are frequency spectrum, power spectrum and power spectrum density (PSD).

3 Configuration of the Device Parameters

3.1 Settings for FFT Analysis

In order to acquire the suitable signals which are available for FFT Analysis the soundcard must be set correctly. The utilization of FFT has also constrains which is called “Nyquist-shannon sampling theorem” or “Sampling theorem”. The constrains are:

1. The sampling rate must be at least twice higher than the band width of signal.
2. The number of samples must be 2^n (n is integer, always the bit number).
3. The sampling period must be the integral multiple of signal period.

The signals which do not fulfill the sampling theorem will cause the aliasing effect and leakage effect during the FFT analysis and generate errors.

3.1.1 The Aliasing Effect

In statistics, signal processing, computer graphics and related disciplines, aliasing refers to an effect that causes different continuous signals to become indistinguishable (or *aliases* of one another) when sampled. It also refers to the distortion or artifact when a signal is sampled and reconstructed as an alias of the original signal. If the sampling rate is not high enough the aliasing effect will be generated. The theory of aliasing generation is illustrated in figure 4.

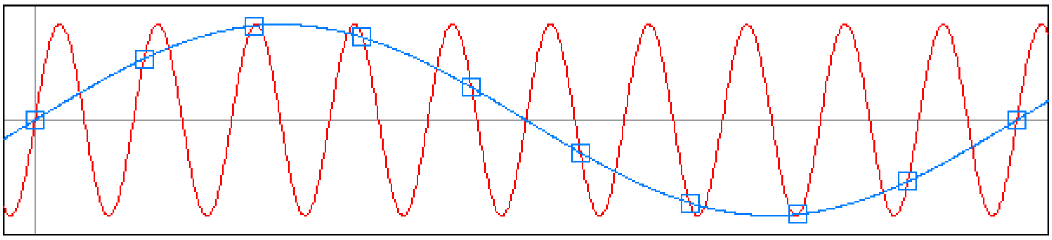

Figure 4 The generation of aliasing effect

In figure 4, it is shown that because of the low sampling rate the two different signals have the same sample values. It can directly influence the frequency spectrum and make it difficult to be recognized [7]. The influence of the aliasing effect in frequency domain is illustrated in figure 5.

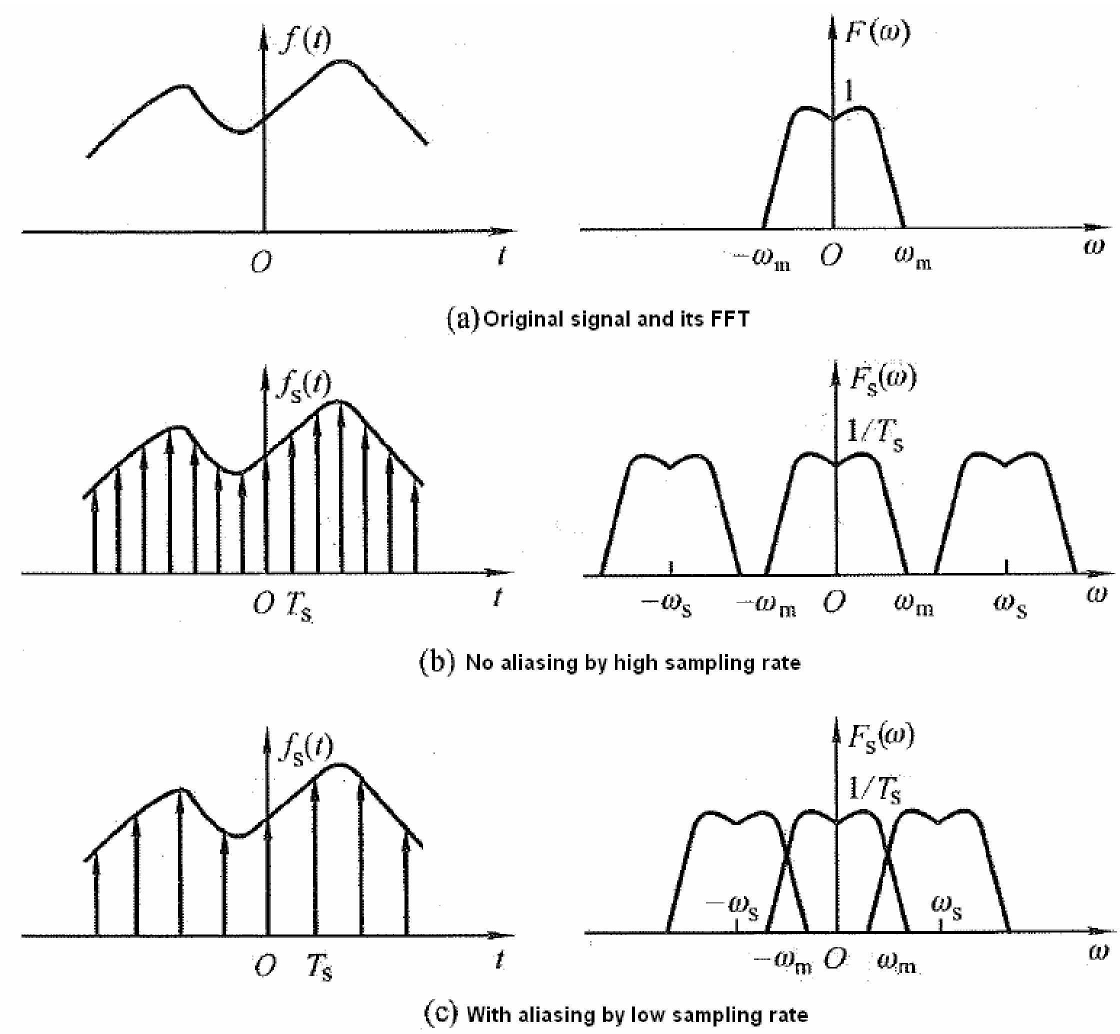

Figure 5 The aliasing in frequency domain and frequency spectrum

Figure 5 shows that because of the aliasing – effect the features which lay in the frequency spectrum are mixed and become more difficult to be recognized. Increasing the sampling rate to be a higher level can avoid or at least reduce this.

3.1.2 The Leakage Effect and Window Functions

The signals are acquired continuously but the available impact acoustic signals are non – continuous because the impacts of particles are discrete. The available signals need to be cut from the unlimited signal series i.e. the time function must be set to be limited in order to analyze the signals. The signal cutting process can be realized by the original signal *x(t)* multiply with a rectangular impulse *h(t).* Just like watching the signal through a rectangular window. The *h(t)* is called window function. The signal after cutting can be calculated as [6]:

$$x_1(t) = x(t)\,h(t) \qquad (1)$$

The Fourier transformation of $x_1(t)$ can be calculated as the convolution of the *X(f)* and *H(f). X(f)* and *H(f)* are the Fourier transformation of *x(t)* and *h(t)* [6]:

$$X_1(t) = X(f) * H(f) \qquad (2)$$

During the cutting process the distortion of the frequency spectrum is generated which is known as the “Leakage effect”. The generation of leakage effect is illustrated in figure 6.

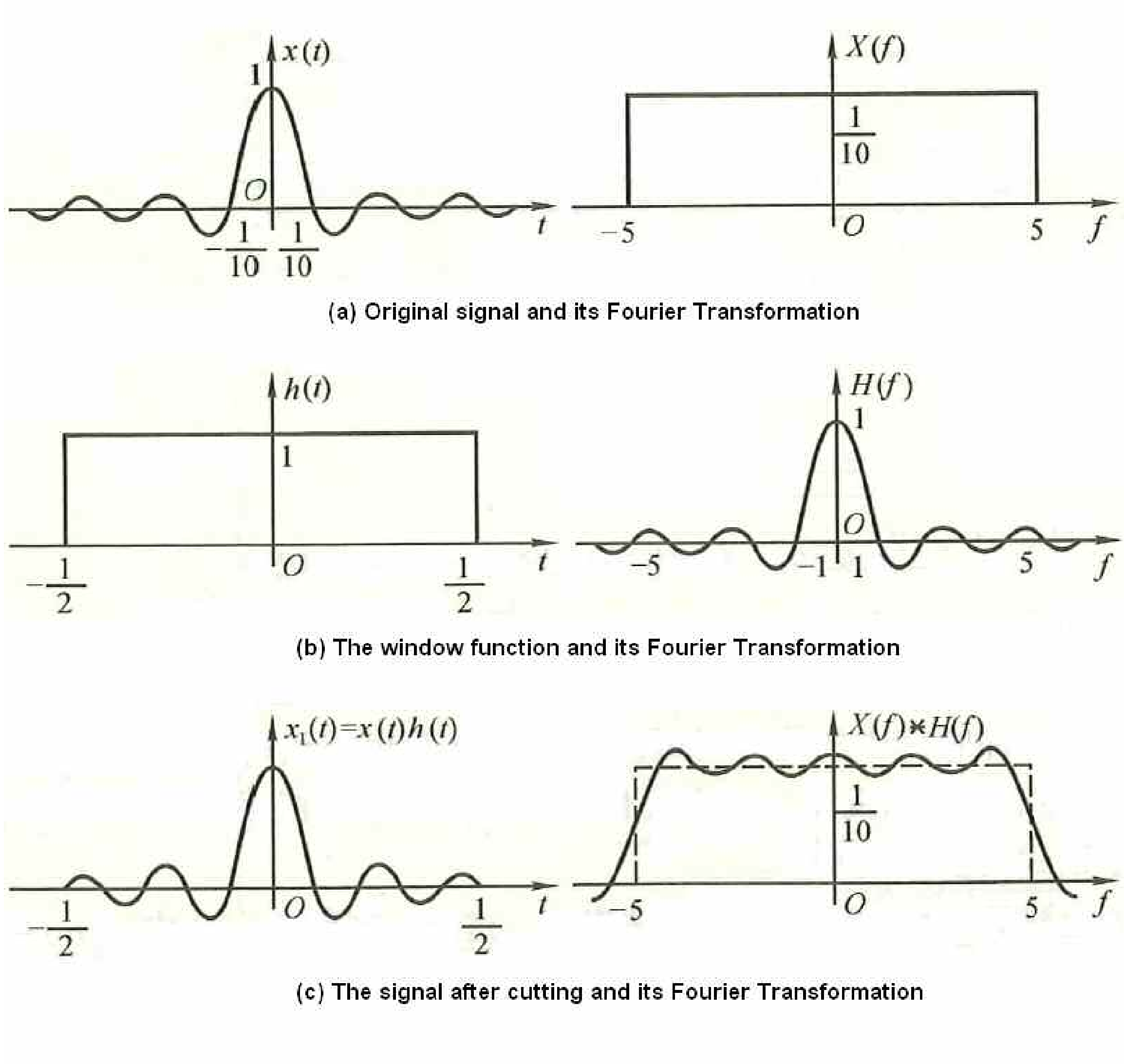

Figure 6 The leakage effect generated by rectangular window function

Figure 6 shows that the Fourier Transformation of the signal which was cut by a window function has the distortions on both band width and frequency spectrum. The band width is expanded and the spectrum has undulations. The expansion of band - width may further cause or intensify the aliasing effect and the spectrum distortion may conceal the information of the signal.

The leakage effect which is generated by truncation of the time – unlimited signals is inevitable. It can only be minimized by a selection of different window functions. The high distortion of the rectangular window is caused by the impulse and mutation in the time domain of this rectangular window function. The impulse in the time domain causes the low convergence in the frequency domain and the solution to this problem is to cut the signal by a gradual change window function. Many of such function have been developed, like the Hanning window, the Hamming window, the Blackman window and etc. These three kinds of window functions are illustrated in figure 7.

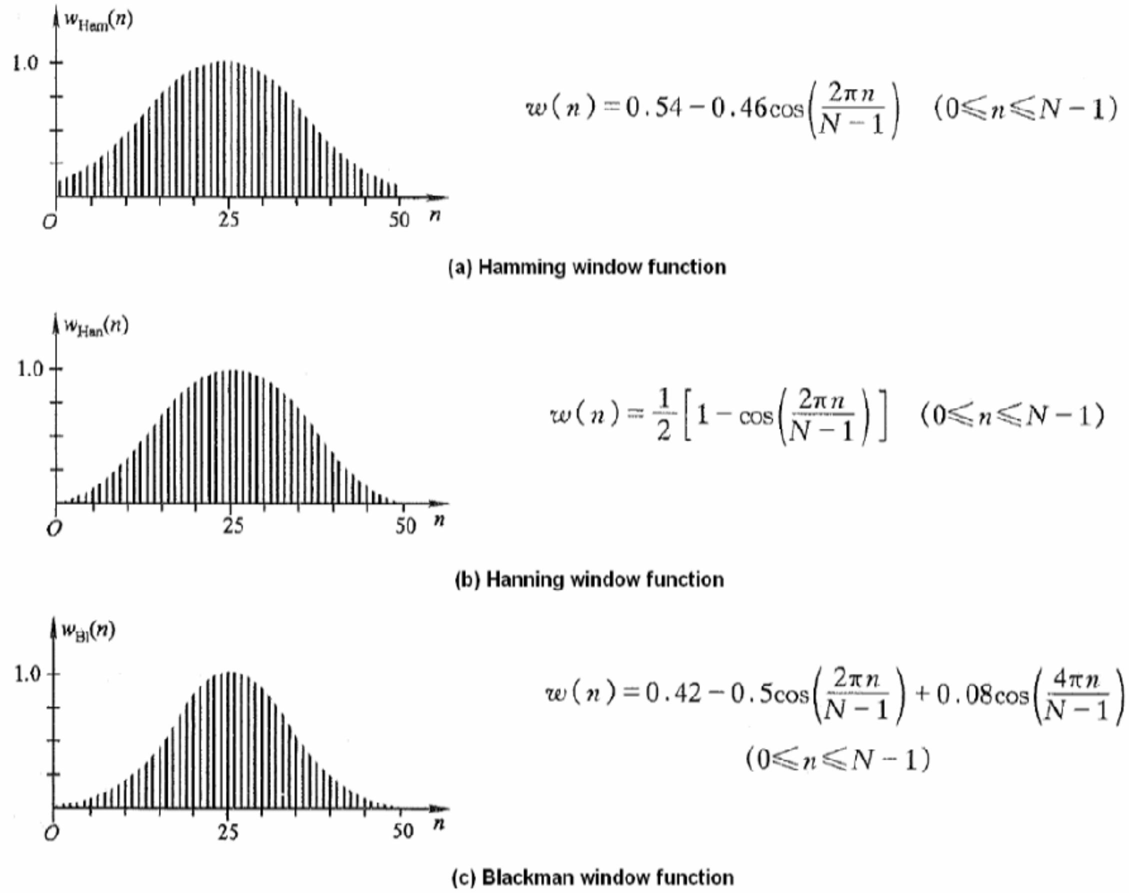

Figure 7 Three kinds of window functions

In these functions *N* represents the width i.e. the number of the samples in a discrete – time window function. Typically it is an integer power – of – 2, such as 2^{16} = 65536. n is an integer with values $0 \le n \le N-1$.

For a given window function, the intensity of the leakage distortion correlates with the side lobe attenuation in its own frequency spectrum. The ideal condition is that the heights of side lobes are zero and all the energy concentrates on the main lobe, which is impossible [6]. In reality the side lobe can only be minimized but not be avoided. For example the rectangular window and the Hanning window are compared in figure 8:

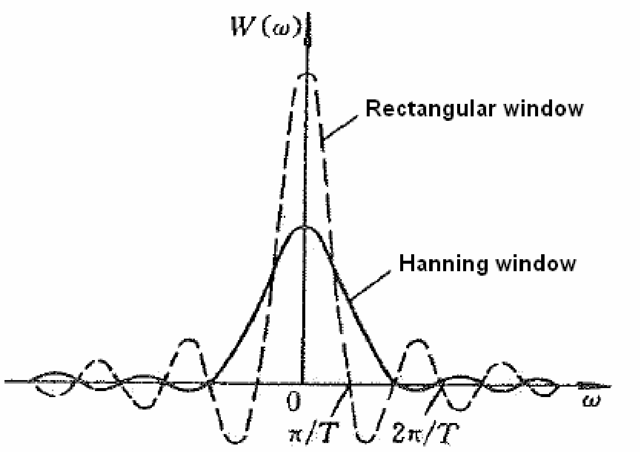

Figure 8 Comparison of Rectangular and Hanning window in frequency domain

It is illustrated that the side lobe attenuation of Hanning window is much more rapid than rectangular window and the corresponding energy leakage is much lower.

By setting of the sampling rate the number of samples and the selection of a suitable window functions with which the acquired acoustic signals can be cut and analyzed correctly, so that the results can also be available to be used as sorting criteria.

One constrain for FFT analysis is that the sampling period must be the multiple integral of a signal period. If the sampling rate and the number of samples are determined, the sampling period is also defined and it does not have to be the multiple integral of a signal period. This problem can cause a small leakage effect and its distortion behaves at the side lobes of the frequency spectrum. This is illustrated in figure 9:

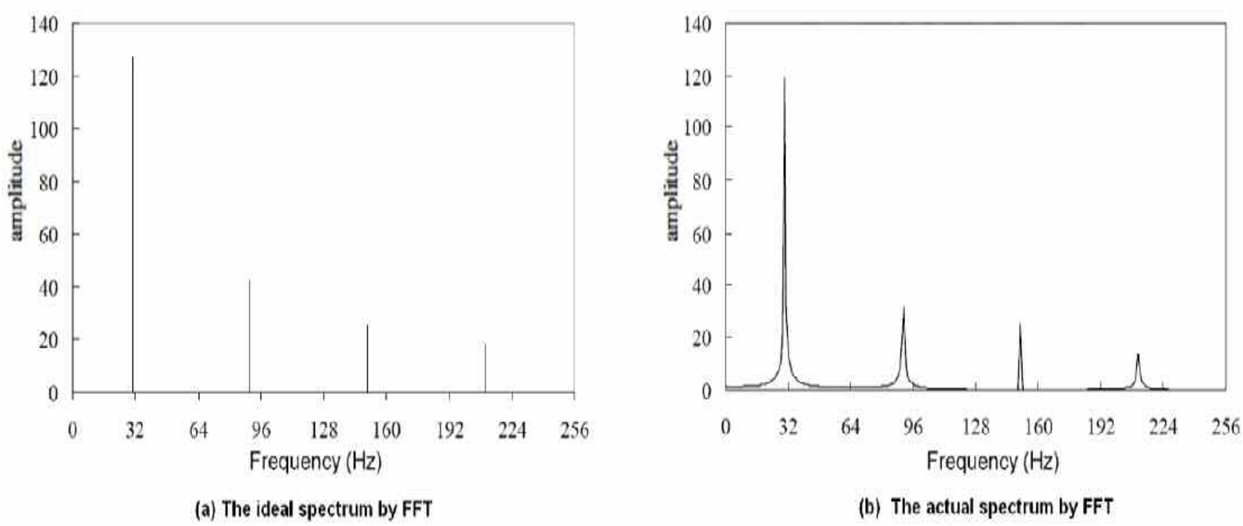

Figure 9 The Comparison of ideal and actual spectrum by FFT Analysis

Figure 9 shows that the ideal eigen – frequencies should be denoted by lines on the FFT spectrum but on the actual spectrum the eigen – frequencies are denoted by peaks because of the side lobes which are generated by wrong sampling periods. Normally this distortion is inevitable and in most cases it does not influence the feature extraction since the energy leakage by expansion of sampling period is very low and the position of eigen – frequencies cannot be aliased or changed. The most important constrains of FFT analysis is the sampling rate and the number of samples.

3.2 Configuration of Soundcard

According to the constrains of the sampling process, the soundcard must be configured before running experiments. The soundcard which is selected for this research has a band width of 0 – 22.05 kHz. Hence the sampling rate is set to be 44.1 kHz. The resolution of the soundcard is set to 16 bit in order to acquire enough information. The setting of the channels is mono. The window function which is selected for cutting the original signal is the "Hanning" window and it is realized by using software.

The setting and control of the sampling process is automatically controlled by LabVIEW software automatically. Using the sound and graphic toolkit one can easily start or close the soundcard and set up all the necessary parameters. The sound signals are set to be saved on the hard disk as ".wav" files, because this format of sound data is nearly universal for all acoustical software and its accuracy is also high enough for industrial measurements. The virtual signal acquirement instrument is illustrated in figure 10.

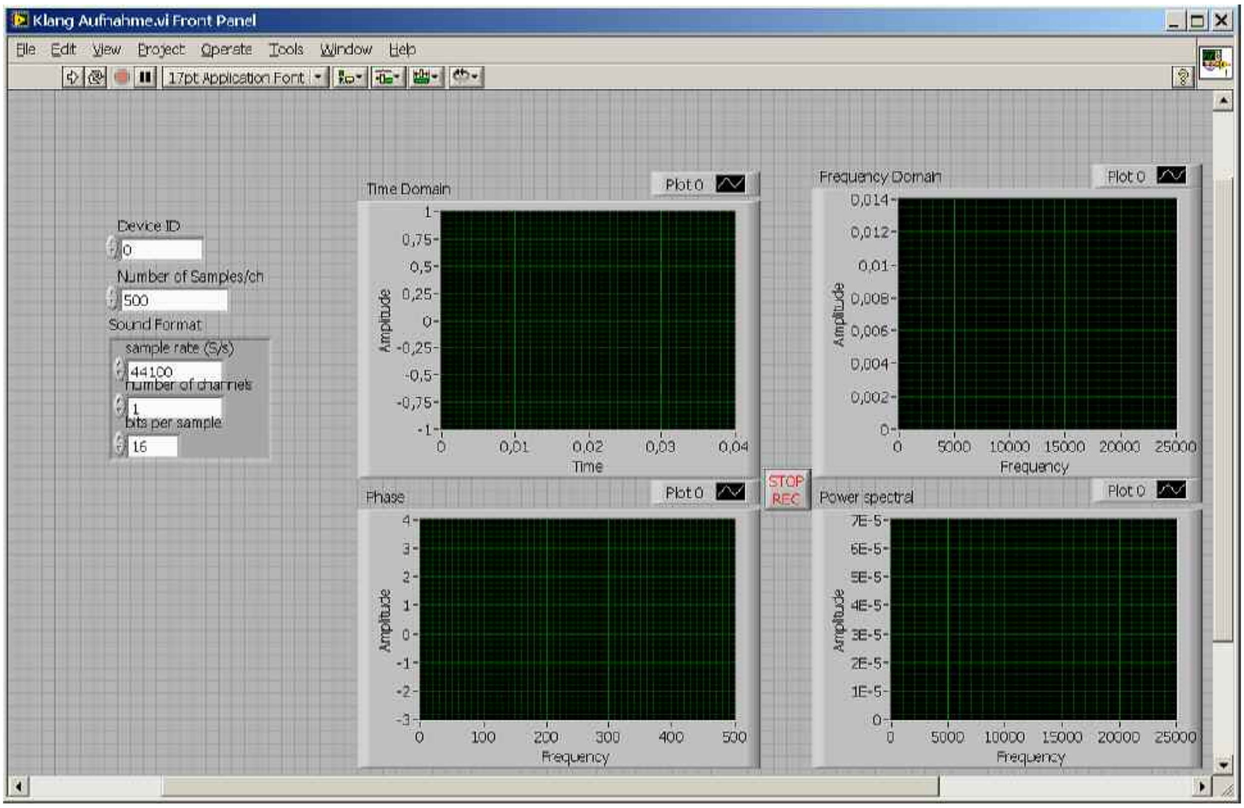

Figure 10 The front panel of signal acquirement program

On this panel, there is only one device (soundcard) installed. So the default device ID is zero. The sampling period is fixed by giving the “number of samples per / channel” and “sampling rate”. The number of channels is one and the resolution is 16 bit. The saving path of signal data is determined in the program but not on the front panel. The waveform, phase and two kinds of spectrums are shown synchronously and continuously. One example of impact acoustic signal from ABS Plastic is illustrated figure 11.

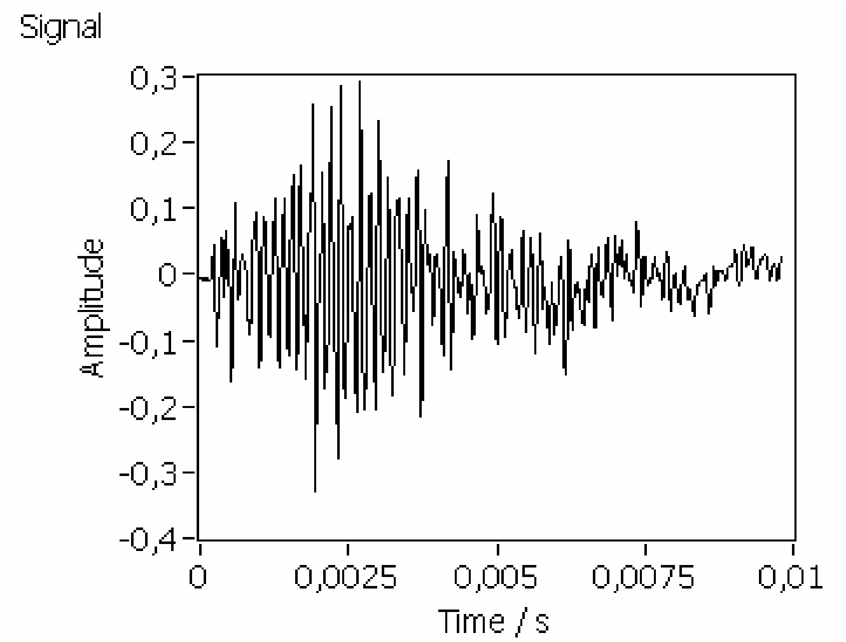

Figure 11 One example of impact acoustic signal from ABS plastic

4 Process of Experiment and Analysis of Results

This paper shows the results from several kinds of acoustic impact emissions from black material particles. The results show that using the FFT analysis the impact characteristics of each material have different features which can be used as sorting criterion. Not all the impacts are available for feature extraction, the abnormal impact such as the double or triple impacts which are caused by the shape of particles and the no – ringing impacts. The abnormal impacts are inevitable but can be minimized by regulating the shape of the particles and increasing the time of single particle impact (circulation of indistinguishable particles).

4.1 Selection of a suitable Shape for Particles

The crushing process of materials can only determine the particle size, the shape of particles are generated randomly. The impact process is also a random process so that the types of acoustic impact emissions of a single particle are different over a certain number of experiments. There could be several kinds of signals by one particle. The abnormal impacts are mostly generated by the irregular shapes of particles. Through this research it can be deduced that by increasing the particle size the probability of abnormal impacts increases. Too small particle on the other hand cannot generate signals with an adequate intensity of the. The suitable particle sizes are between 5 mm and 25 mm. The abnormal impacts which are caused by particle shapes are illustrated in figure 12.

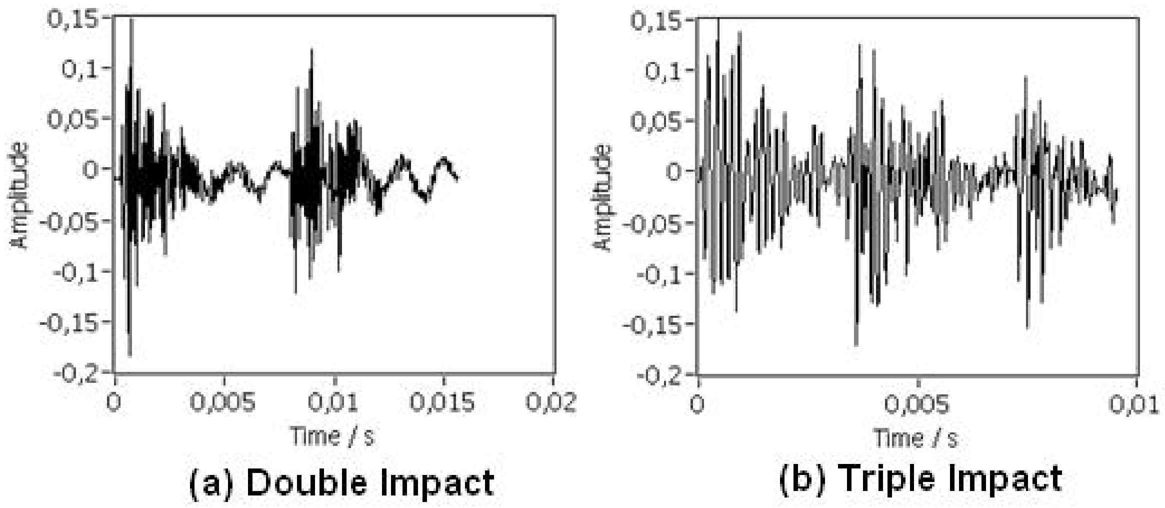

Figure 12 The abnormal Impacts generated by Particle Formats

The influence of multi – impact in frequency domain by FFT is the increasing of spikes of spectrums and further the decentralization of the features. For example the power spectrum of the signal which is shown in figure 12 (a) is shown in figure 13:

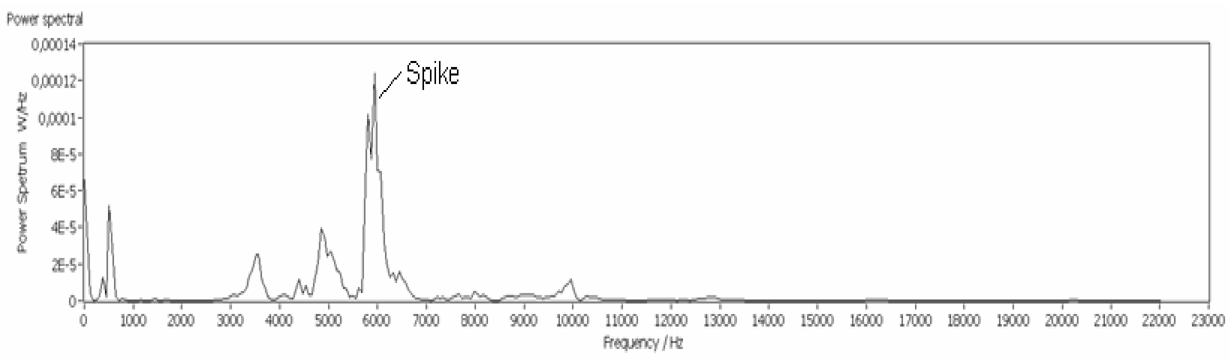

Figure 13 The influence of multi – impact on Power Spectrum

The feature of the signal in figure 12 (a) should have the highest peak in the range of 5,700 – 6,000 Hz. Due to the abnormal impact there are two peaks in this range and make this spectrum similar to the one of another material. This problem can confuse the program and lead to a wrong determination in sorting.

Except the multi – impact sometimes other abnormal impacts may occur. This refers to impacts without ringing. Sounds induced by impact can be separated into two major categories. The first is “acceleration” and the second is “ringing”. The acceleration component of the process controls the early time response of the time-dependent field pressures, whereas the subsequent time response is dominated by the free-vibration of the impacting bodies. Ringing sounds which control the response after the decay of the acceleration component is traditionally recognized as useful for the feature extraction [3]. The acceleration and ringing parts of sound signals are illustrated in figure 14.

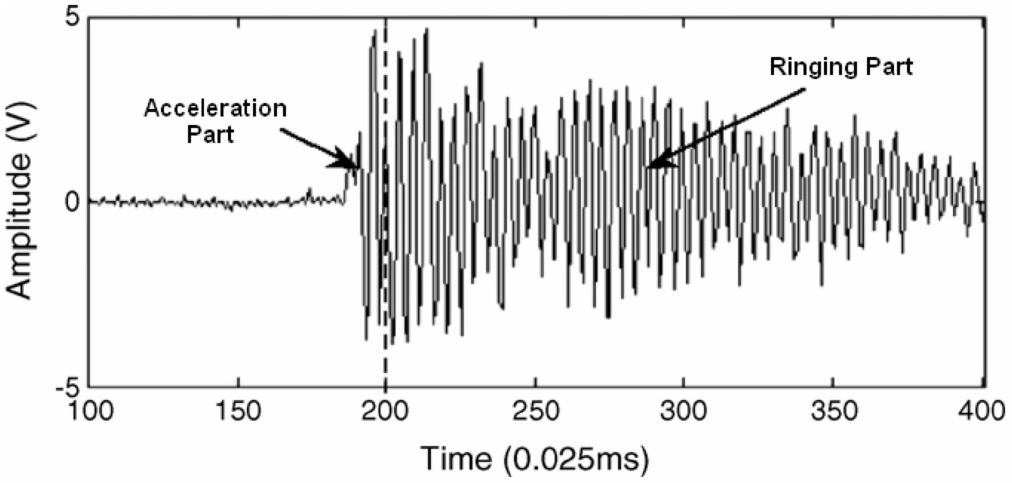

Figure 14 The acceleration and ringing parts of a sound signal

Moreover, the initial acceleration part of impact sounds can be further divided into two parts, from the impact plate and from the test object. If the particles are not correctly accelerated to vibrate, the ringing part will be very weak and most of the impact energy will be transferred to the impact plate. The result of the analysis will be the vibration spectrum of the impact plate. This kind of abnormal impact happens infrequently very seldom but cannot be ignored. The particles which impact the plate without ringing signal must also be determined and sent back to the raw material and impact again.

According to the research of all particle sizes the probability of multi – impact is about max. 30% and the probability of the impacts without ringing are about max. 5%. By using the particle with the size of 5 mm – 25 mm the multi – impacts can be reduced to 3% - 10% and the impacts without ringing can be reduced to about 2%.

4.2 The Feature of the Impact Plate

The energy distribution during the impact is also a random process. In which a part of the kinetic energy of the falling particles is transferred to the impact plate and causes it to vibrate. The vibration of the impact plate influences the spectrum of the acquired signal and leads to distortions. Hence the response feature of the impact plate during the impact process must be determined and then neglected during the FFT analysis.

The material, shape, size and thickness of the impact plate are fixed so that the response features of it should also be a fix value. The response features of the impact plate are determined by the analysis of the impact signals which are generated by different particles with different materials. The analysis was done by the sound processing software DEWESoft. The results are illustrated in figure 15:

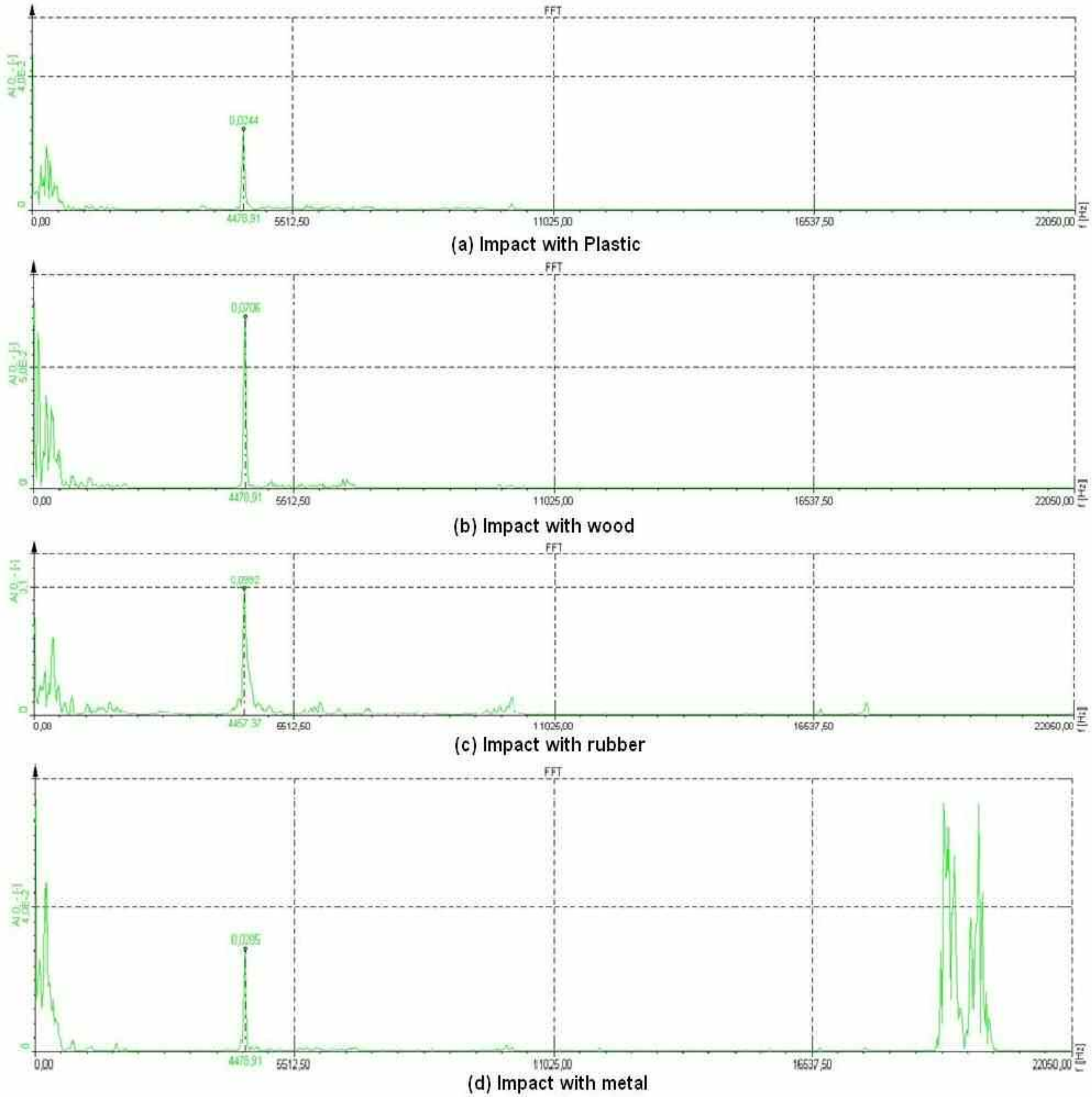

Figure 15 The FFT analysis of the eigen – frequencies of impact plate

In figure 15 the impact signal from different materials are analyzed and the results shown that the common feature are the peaks between 4,457.37 Hz and 4,478.91 Hz. In order to avoid distortions the peaks between 4,400 Hz and 4,600 Hz on the spectrum of signals can be ignored as the features of the impact plate.

4.3 Analysis of the obtained Results

In the time domain, each signal which is cut from the signal series contains about 500 points sampled at 44.1 kHz. To obtain the frequency – domain information, the power spectral of the feature is estimated with the Fast Fourier Transformation (FFT) calculation on the original time history.

The particles which are used for the experiment have the same particle size about 20 mm and the same thickness of about 2.5 – 3 mm. The plastic flakes are usually mixed together and then crushed into the same particle size. The particle size of 20 mm is suitable to avoid the abnormal impacts. In this experiment there are 3 different plastics in total which have been measured for min. 150 times and they all have one or more evident features in the frequency domain.

The 3 kinds of plastics are Polypropylene (PP), Styrene maleic anhydride (SMA) and Acrylonitrile butadiene styrene (ABS). The typical spectrums have been obtained and illustrated in figure 16 to 18.

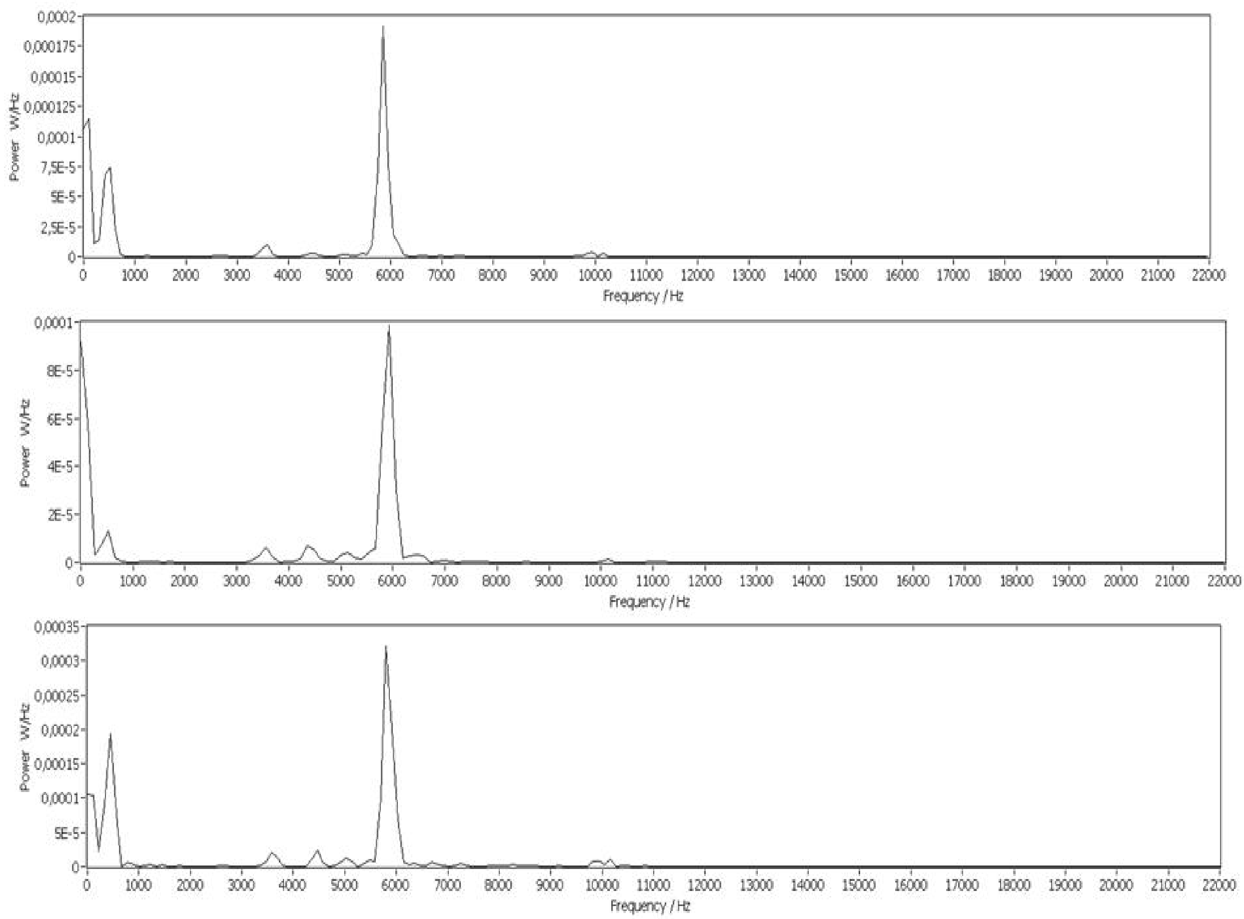

Figure 16 The power spectrum of Polypropylene (PP) pieces

Figure 16 illustrates that the spectral feature peaks of PP particles concentrate in the range of 5,700 – 6,000 Hz. Normally the main peak has not spikes and it is at least 8 times higher than other peaks. The peaks which locate in the range of 4,400 – 4,600 Hz must be ignored because it is the feature area of the impact plate. This feature has the probability of occurrence of about 62.5%.

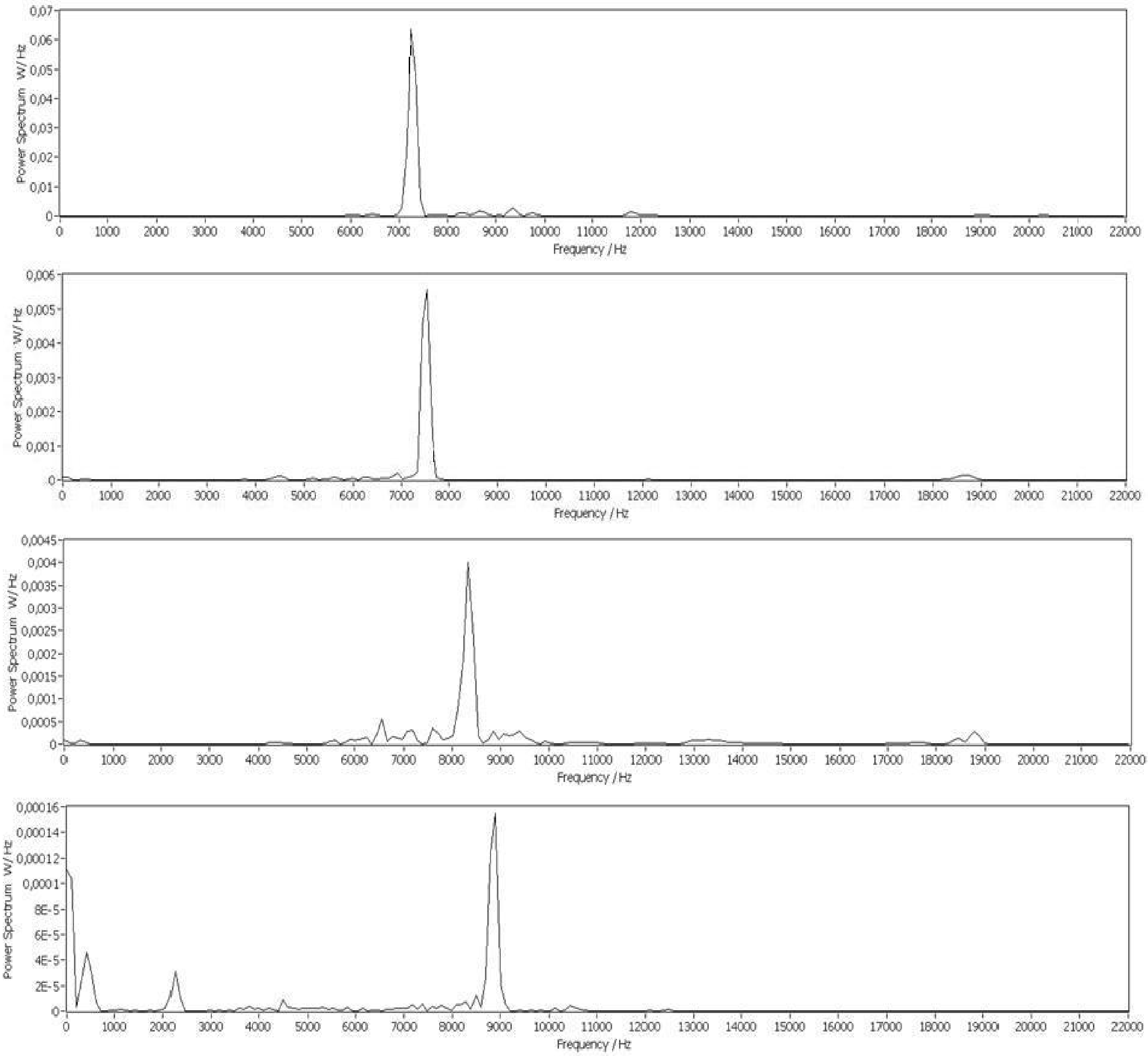

Figure 17 The power spectrum of Acrylonitrile butadiene styrene (ABS) pieces

Figure 17 illustrates that the spectral features of ABS particles are distributed in the range from 7,000 to 9,000 Hz. Normally the main peaks are smooth and do not have spikes. Sometimes there is not only one main peak but also the second or third high peaks on the spectrum. The main high peaks are always located in the range of 6,000 – 6,500 Hz and 7,000 – 9,000 Hz. The approximate probabilities of occurrence for the main peak areas are:

1. 7,000 – 8,000 Hz: 35%
2. 8,000 – 9,000 Hz: 25%
3. 6,000 – 6,500 Hz: 10%

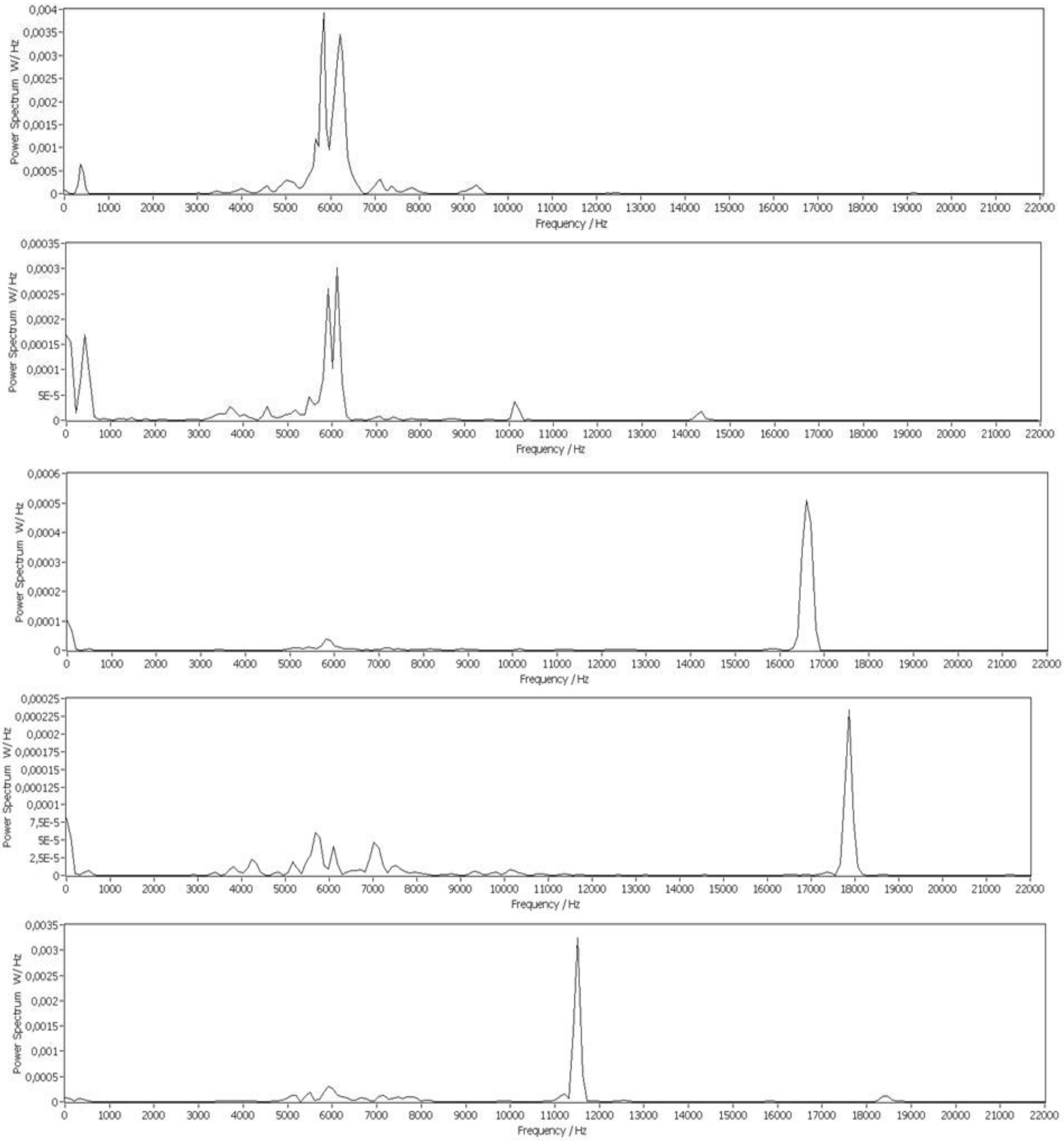

Figure 18 The power spectrum of Styrene maleic anhydride (SMA) pieces

Figure 18 illustrates that the spectral features of SMA pieces are more complicated than the others. There are two different cases:

1. There are double peaks locate respectively in the range of 5,700 – 6,000 Hz and 6,000 – 6,300 Hz. The valley which locates between the double peaks is in the range of 5,900– 6,100 Hz. In the range of > 10,000 Hz there are two or three small peaks. This case has a probability of occurrence about 30%.
2. The highest peak locates in the wide range of 11,000 – 21,000 Hz. Sometimes this range still contains another two or three high peaks. The common ground of

this kind of spectrum is that there is a small peak located in the range of 5,700 – 6,100 Hz. This case has a probability of occurrence about 30%.

5 Conclusion

In this paper the basic principles and concept for acoustic sorting technologies are introduced and the methods for acquisition and analysis of acoustic impact signals are also given, by using the soundcard and computer a signal DAQ and the analysis system can be easily established and through the methods based on the Fast Fourier Transformation the features of most impacts can be deduced and used in further as application as sorting criteria.

In this paper the equipment for acoustical research was introduced and actual experiments carried out with 3 different kinds of black materials have been summarized and the results illustrate that the features which are extracted from the impact signals are evident and can be easily distinguished from one another. In further work, the results could be used for technical applications. A real experimental system of acoustic sorting can be built based on this research.

6 Literature

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